AMENDMENTS TO THE CLAIMS

1. (Currently amended) A method of separating a desired speech signal in an acoustic environment, comprising:

receiving a plurality of input signals, the input signals being generated responsive to the desired speech signal and other acoustic signals;

processing the received input signals using an independent component analysis (ICA) or blind source separation (BSS) method under stability constraints, wherein the ICA or BSS method modulates the mathematical formulation of mutual information directly or indirectly through approximations; and

separating the received input signals into output channels comprising ~~at~~ one or more desired audio output signals and one or more noise output signals.

2. (Original) The method according to claim 1, wherein one of the desired audio signals is the desired speech signal.

3. (Currently amended) The method according to claim 1, wherein ~~where~~ the ICA ~~or BSS~~ method further comprises ~~process includes~~ minimizing or maximizing the mathematical formulation of mutual information directly or indirectly through approximations.

4. (Currently amended) The method according to claim 1, wherein the stability constraints comprise ~~further comprising the step of stabilizing the ICA process by~~ pacing the adapting of an ICA filter ~~weight adaptation dynamics~~.

5. (Currently amended) The method according to claim 1, wherein the stability constraints comprise ~~further comprising the step of stabilizing the ICA process by~~ scaling the received input signals ~~ICA inputs~~ using an adaptive scaling factor, the adaptive scaling factor being selected to constrain weight adaptation speed.

6. (Currently amended) The method according to claim 1, wherein the stability constraints comprise ~~further comprising the step of stabilizing the ICA process by~~ filtering learned filter weights in the time domain and the frequency domain, the filtering selected to avoid introduction of artificial reverberation effects.

7. (Currently amended) The method according to claim 1, ~~wherein~~ further comprising applying peripheral pre-processing or post-processing techniques ~~are applied~~ to at least one of the received input signals ~~and~~ or at least one of the separated output signals ~~in varying degrees~~.

8. (Currently amended) The method according to claim 1, further comprising ~~utilizing~~ pre-processing the received input signals ~~techniques or information to enhance the performance of the separation~~.

9. (Original) The method according to claim 8, further comprising improving the conditioning of a mixing scenario applied to the input signals.

10. (Currently amended) The method according to claim 2, further comprising utilizing characteristic information of the desired speech signal to identify the output channel containing the separated desired speech signal.

11. (Original) The method according to claim 10 wherein the characteristic information is spatial, spectral or temporal information.

12. (Currently amended) The method according to claim 1, further comprising applying ~~wherein~~ post-processing techniques to at least one of the separated output signals using ~~are used to improve the quality of the desired signal utilizing the~~ at least one processing signal selected from ~~at least~~ one or more of the noise signals ~~or~~ and one or more ~~at least one~~ of the input signals.

13. (Currently amended) The method according to claim 12, wherein the using at least one processing signal consists of using the ~~post-processing techniques further include including the step of using the separated~~ noise signal ~~to further separate and enhance the desired speech signal~~.

14. (Currently amended) The method according to claim 13 wherein the using the noise signal comprises ~~step includes~~ using the noise signal to estimate the noise spectrum for a noise filter.

15. (Currently amended), The method according to claim 1, further comprising: ~~further including~~

spacing apart at least a first and a second microphone; and

generating one of the input signals at each respective microphone.

16. (Currently amended) The method according to claim 15, wherein the spacing apart at least a first and a second microphone comprises ~~step includes~~ spacing the microphones between about 1 millimeter and about 1 meter ~~1mm and about 1m~~ apart.

17. (Currently amended) The method according to claim 15, wherein the spacing apart at least a first and a second microphone comprises ~~step includes~~ spacing the microphones

apart on a telephone receiver, a headset, or a hands-free kit.

18. (Currently amended) The method according to claim 1, wherein the ICA ~~process~~ or BSS method comprises:

adapting a first adaptive ICA filter connected to a first output signal and to a second input signal~~, the first filter being adapted~~ by a recursive learning rule involving the application of a nonlinear bounded sign function to ~~the noise signal channel~~ one or more noise output signals; and

adapting a second adaptive ICA filter connected to a first input signal and to a second output signal~~,, the second filter being adapted~~ by a recursive learning rule involving the application of a nonlinear bounded sign function to the one or more desired ~~speech signal channel~~ audio output signals,

wherein the first filter and the second filter are repeatedly applied to produce the desired speech signal.

19. (Currently amended) The method according to claim 18, ~~wherein (a)~~ further comprising:

spacing apart at least a first and a second microphone;

generating one of the input signals at each respective microphone;

recursively filtering the one or more desired audio output signals ~~speech signals channel recursively filtered~~ by the first adaptive independent component analysis filter to obtain a recursively filtered speech signal;

~~is fed back and added to the input channel from the second microphone, thereby producing the noise signal channel, and (b)~~ recursively filtering the one or more noise output signals ~~signal channel recursively filtered~~ by the second adaptive independent component analysis filter to obtain a recursively filtered noise signal;

adding the recursively filtered speech signal to the input signal from the second microphone, thereby producing the noise output signal; and

adding the recursively filtered noise signal ~~is fed back and added~~ to the input ~~channel~~ signal from the first microphone, thereby producing the one or more desired audio output signals ~~speech signal channel~~.

20. (Currently amended) The method according to claim 19, wherein the received input ~~channel~~ signals are inversely scaled ~~down~~ by an adaptive scaling factor computed from a

recursive equation as a function of ~~the~~ incoming signal energy.

21. (Currently amended) The method according to claim 1 ~~claim 18~~, further comprising:

stabilizing a ~~wherein the~~ ~~filter weight~~ recursive learning rule adapting ~~for~~ the first adaptive ICA ~~cross~~ filter ~~is stabilized~~ by smoothing ~~the filter~~ coefficients of the first adaptive ICA filter in time;~~,~~ and

stabilizing a ~~wherein the~~ ~~filter weight~~ recursive learning rule adapting ~~for~~ the second adaptive ICA ~~cross~~ filter ~~is stabilized~~ by smoothing ~~the filter~~ coefficients of the second adaptive ICA filter in time.

22. (Currently amended) The method according to claim 1 ~~claim 18~~, wherein the filter weights of the first adaptive ICA cross ~~filter weights~~ are filtered in the frequency domain, and wherein the filter weights of the second adaptive ICA cross ~~filter weights~~ are filtered in the frequency domain.

23. (Currently amended) The method according to claim 1 ~~claim 18~~, further comprising post processing ~~a post processing module connected to~~ the desired speech signal comprising ~~which applies a single or multi channel speech enhancement module including~~ voice activity detection and wherein ~~the~~ post-processed outputs are not fed back to ~~the~~ input signals ~~channels~~.

24. (Currently amended) The method according to claim 1, ~~claim 18~~ wherein the ICA method ~~process~~ is implemented in a fixed point computing precision environment and wherein the ICA method further comprises:

applying ~~where~~ the adaptive ICA ~~cross~~ filters ~~are applied~~ at every sampling instant;~~,~~

updating ~~but where~~ filter coefficients ~~are updated~~ at multiples of the sampling instant;~~,~~ and

adapting filter lengths of variable sizes ~~size are~~ according to ~~used to fit~~ the computational power available.

25. (Currently amended) The method according to claim 1 ~~claim 18~~, further comprising applying spectral subtracting to ~~post processing~~ the one or more desired audio output signals ~~speech signal using the noise signal, the post processing module applying spectral subtraction to the desired speech signal~~ based on the one or more noise signals ~~signal~~.

26. (Currently amended) The method according to claim 1 ~~claim 18~~, further comprising applying Wiener filtering to ~~post processing~~ the one or more desired audio output signals ~~speech signal using the noise signal, the post processing module applying spectral subtraction to the desired speech signal~~ based on the one or more noise signals ~~signal~~.

27. (Currently amended) The method according to claim 1 ~~claim 18~~, further generating ~~comprising receiving~~ a third set of audio input signals from at a third microphone ~~channel~~, and applying a nonlinear bounded function to incoming signals using a third filter.

28. (Canceled)

29. (Canceled)

30. (Canceled)

31. (Canceled)

32. (Canceled)

33. (Canceled)

34. (Canceled)

35. (Canceled)

36. (Canceled)

37. (Canceled)

38. (Canceled)

39. (Currently amended) A system for separating a desired speech signal ~~signals~~ in an acoustic environment, comprising

a plurality of input channels each receiving one or more acoustic signals, wherein the one or more acoustic signals comprises a speech signal;

at least one independent component analysis (ICA) or blind-source separation (BSS) ~~ICA or BSS~~ filter module comprising an ICA or BSS filter that~~, wherein the filter~~ separates the received signals ~~under stability constraints~~ into one or more desired audio signals and one or more noise signals;

a stability constraint, wherein the stability constraint at least partially stabilizes the at least one ICA or BSS filter; and

a plurality of output channels transmitting the separated signals,

wherein the filter modulates the mathematical formulation of mutual information directly or indirectly through approximations.

40. (Currently amended) The system according to claim 39, wherein the one or more acoustic signals comprise the one or more desired audio signals ~~signal is a speech signal received in the plurality of acoustic signals~~.

41. (Canceled)

42. (Currently amended) The system according to claim 39, wherein implementing the stability constraint ~~filter stabilizes the ICA process by~~ ~~pacing~~ paces adaptation of the ICA or BSS filter ~~weight adaptation dynamics~~.

43. (Currently amended) The system according to claim 39, wherein implementing the stability constraint comprises ~~filter stabilizes the ICA process by~~ scaling ICA or BSS inputs using an adaptive scaling factor, the adaptive scaling factor selected to constrain ~~weight~~ adaptation speed.

44. (Currently amended) The system according to claim 39, wherein implementing the stability constraint comprises ~~filter stabilizes the ICA process by~~ filtering learned filter weights in the time domain and the frequency domain, the filter selected to avoid introduction of artificial reverberation effects.

45. (Currently amended) The system according to claim 39, further comprising one or more processing modules comprising at least one filter selected from a pre-processing peripheral filter and a post-processing peripheral ~~processing~~ filter ~~filters~~ applied to the one or more acoustic signals ~~input~~ and/or the separated ~~output~~ signals.

46. (Currently amended) The system according to claim 45, wherein the filter is the ~~further comprising one or more~~ pre-processing peripheral filter ~~filters~~.

47. (Currently amended) The system according to claim 45, wherein the filter is the ~~further comprising one or more~~ post-processing peripheral filter ~~filters~~.

48. (Currently amended) The system according to claim 39, further comprising one or more microphones connected to the plurality of input channels.

49. (Currently amended) The system according to claim 48, wherein the one or more microphones are two or more microphones, and wherein ~~comprising two or more microphones~~ each of the two or more microphones is spaced ~~apart~~ between about 1 millimeter and about 1 meter ~~1mm and about 1m~~ apart.

50. (Original) The system according to claim 39, wherein the system is constructed on a hand-held device.

51. (Currently amended) The system according to claim 39, wherein the at least one ICA or BSS filter module comprises ~~includes~~:

a first adaptive independent component analysis (ICA) filter connected to a first output channel and to a second input channel, the first filter being adapted by a recursive learning rule involving the application of a nonlinear bounded sign function to the one or more noise signals ~~signal channel~~;

a second adaptive independent component analysis filter connected to a first output channel and to a second input channel, the second filter being adapted by a recursive learning rule involving the application of a nonlinear bounded sign function to the desired speech signal ~~channel~~;

wherein the first filter and the second filter are repeatedly applied to produce the desired speech signal.

52. (Canceled)

53. (Canceled)

54. (Canceled)

55. (New) The system according to claim 39, wherein the plurality of input channels comprises at least two spaced-apart microphones constructed to receive the acoustic signals, the microphones being an expected distance from a speech source;

wherein the at least one ICA or BSS filter module is coupled to the microphones; and

wherein the at least one ICA or BSS filter module is configured to:

receive sound signals from the two microphones; and

separate the sound signals under the stability constraint into at least one desired speech output signal line and at least one noise output signal line.

56. (New) The system according to claim 55, further comprising a post-processing filter coupled to the noise output signal and to the desired speech output signal.

57. (New) The system according to claim 55, wherein the microphones are spaced about 1 millimeter to about 1 meter apart.

58. (New) The system according to claim 57 further comprising a pre-processing module configured to pre-process the acoustic sound signals received at each microphone.

59. (New) The system according to claim 55, wherein one of the microphones is on a face of a device housing and the other microphone is on another face of the device housing.

60. (New) The system according to claim 55, wherein the system is integrated into a speech device.

61. (New) The system according to claim 60, wherein the speech device comprises a wireless phone.

62. (New) The system according to claim 60, wherein the speech device comprises a hands-free car kit.

63. (New) The system according to claim 60, wherein the speech device comprises a headset.

64. (New) The system according to claim 60, wherein the speech device comprises a personal data assistant.

65. (New) The system according to claim 60, wherein the speech device comprises a handheld bar-code scanning device.